What is claimed is:

1. A voice coding method based on analysis-by-synthesis vector quantization using a code book containing a voice source code vector having only a plurality of non-zero amplitude values, comprising the step of

variably controlling a position of a sample of the non-zero amplitude value using an index and a transmission parameter indicating a feature amount of voice.

2. The method according to claim 1, further comprising the step of

variably controlling the position of the sample of the non-zero amplitude value using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

3. The method according to claim 2, further comprising the step of

reconstructing the position of the sample of the non-zero amplitude value within a region corresponding to the lag value depending on a relationship between

the lag value and a frame length which is a coding unit of the voice.

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4. The method according to claim 1, further comprising the step of

variably controlling the position of the sample of the non-zero amplitude value using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

5. The method according to claim 4, further comprising the step of

reconstructing the position of the sample of the non-zero amplitude value within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a coding unit of the voice.

6. The method according to claim 5, further comprising the step of

reconstructing the position of the sample of the non-zero amplitude value within a region corresponding to the lag value depending on the pitch gain value.

7. A voice decoding method for decoding a voice signal coded by a voice coding method based on analysis-by-synthesis vector quantization using a code book containing a voice source code vector having only a plurality of non-zero amplitude values, comprising the step of

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variably controlling a position of a sample of the non-zero amplitude value using an index and a transmission parameter indicating a feature amount of voice.

8. The method according to claim 7, further comprising the step of

variably controlling the position of the sample of the non-zero amplitude value using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

9. The method according to claim 8, further comprising the step of

reconstructing the position of the sample of the non-zero amplitude value within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a coding

unit of the voice.

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10. The method according to claim 7, further comprising the step of

variably controlling the position of the sample of the non-zero amplitude value using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

11. The method according to claim 10, further comprising the step of

reconstructing the position of the sample of the non-zero amplitude value within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a coding unit of the voice.

12. The method according to claim 11, further comprising the step of

reconstructing the position of the sample of the non-zero amplitude value within a region corresponding to the lag value depending on the pitch gain value.

13. A voice coding apparatus based on analysis-by-

synthesis vector quantization using a code book containing a voice source code vector having only a plurality of non-zero amplitude values, comprising

a configuration variable code book unit variably controlling a position of a sample of the non-zero amplitude value using an index and a transmission parameter indicating a feature amount of voice.

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14. The apparatus according to claim 13, wherein

said configuration variable code book unit variably controls the position of the sample of the non-zero amplitude value using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

15. The apparatus according to claim 13, wherein

said configuration variable code book unit variably controls the position of the sample of the non-zero amplitude value using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

16. A voice decoding apparatus for decoding a voice

signal coded by a voice coding apparatus based on analysis-by-synthesis vector quantization using a code book containing a voice source code vector having only a plurality of non-zero amplitude values, comprising

a configuration variable code book unit variably controlling a position of a sample of the non-zero amplitude value using an index and a transmission parameter indicating a feature amount of voice.

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17. The apparatus according to claim 16, wherein

said configuration variable code book unit variably controls the position of the sample of the non-zero amplitude value using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

18. The apparatus according to claim 16, wherein

said configuration variable code book unit variably controls the position of the sample of the non-zero amplitude value using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.